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REPORT

DIGITAL SOUND SIGNALS: tests to compare the performance of five companding systems for high-quality sound signals

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Research Department, Engineering Division
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Summary

A number of digital companding systems have been proposed for saving bit-rate in digital sound transmission. All of these systems introduce a degree of programme-modulated noise into the sound signal, and this Report describes tests which have been carried out to compare the resulting subjective impairment. The performance of five companding systems was assessed and, as a result, a decision was made to use a near-instantaneous companding system known as NICAM-3 for future BBC sound-programme links.

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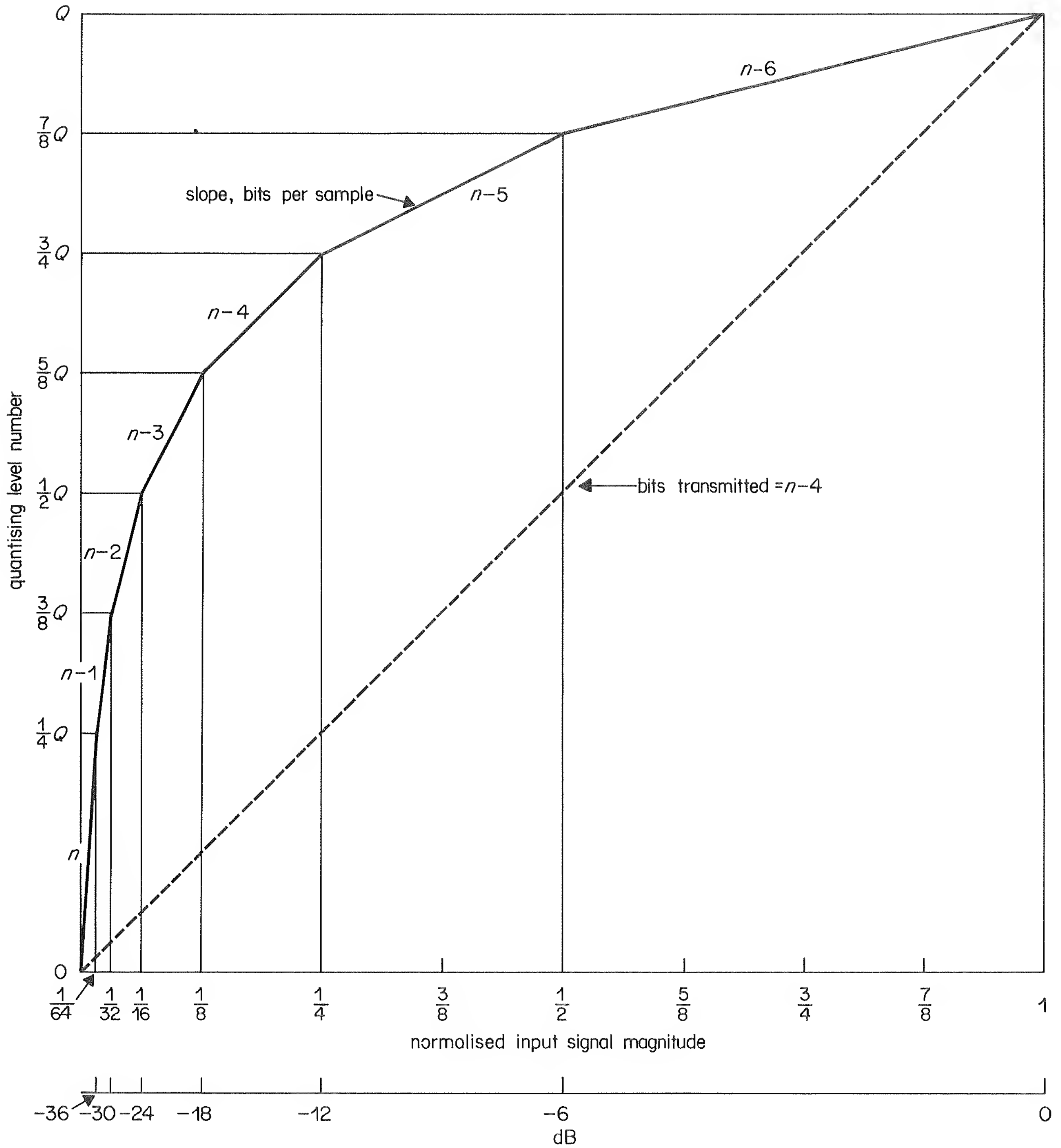

Fig. 1 - 7-segment A companding law: positive quadrant only shown
Q represents the maximum number of digital codes available in positive quadrant; Q is a power of two

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1. Introduction

In order to increase the programme-carrying capacity of pulse-code modulation (p.c.m.) sound links, a number of companding (compressing and expanding) techniques for bit-rate reduction of p.c.m. sound signals have been devised. All of the systems considered in this Report operate by coding high-amplitude sound signals to a lower accuracy than signals with a small amplitude. This reduction in accuracy for high-amplitude signals means that a shorter data word can be used to describe each sample of the sampled sound signal. As a result of the companding operation, the quantising noise present with high-amplitude signals is at a higher level than with low-amplitude signals.* However, the signal itself tends to conceal quantising noise, particularly if pre-emphasis is employed. This concealment needs to be effective, otherwise the programme-modulated noise may be obtrusive to the listener.

Four companding systems were given extensive tests and this Report describes the work involved and presents the results obtained. A fifth system, evolved during the course of the tests, was also tested. Results for this system are included in this Report.

2. Brief description of the companding systems

The four companding systems which were involved in the initial tests were the Italian A-law system from RAI (Radiotelevisione Italiana), the French 9-bit system from TDF (Telediffusion de France), the British NICAM** system developed by the BBC and a new BBC NICAM system (NICAM-2) which would have certain advantages over the original NICAM (which is now referred to as 'NICAM-1'). All systems have a sampling frequency of 32 kHz and employ CCITT pre- and de-emphasis to reduce audibility of programme-modulated quantising noise.

The A-law system uses an initial 14-bit linear coding, with compression to 10 bits. Compression is achieved by retaining the closely spaced quantising levels for low-level signals, but making the quantum steps progressively larger towards the extremities of the coding range according to a fixed law (the A-law). Fig. 1 shows the positive quadrant of the A-law compression characteristic, which comprises a number of straight-line segments. The complementary expansion characteristic is embodied in the decoder. Since these two processes are carried out in the digital domain there is a precise match between compression and expansion characteristics.

* Analogue companding systems also suffer from programme-modulated noise, although in this case the noise is the random noise associated with linear circuits.

** 'Near-instantaneous companding audio multiplex'.

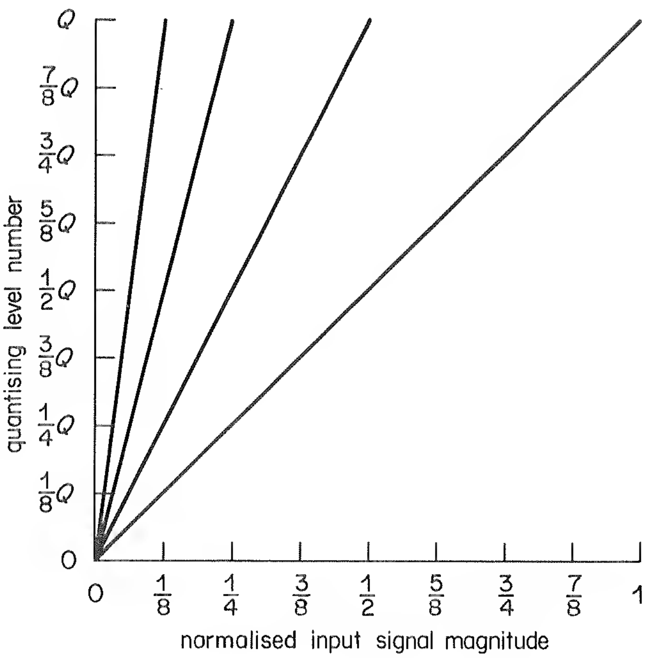

Fig. 2 - Near-instantaneous 4-segment companding law: positive quadrant only shown. Q represents the maximum number of digital codes available in the positive quadrant

The A-law system is the only example of 'instantaneous' companding that was tested. The effective compression and expansion of the signal in such a system is determined by the instantaneous amplitude of each signal sample.

An alternative technique, that of 'near-instantaneous' companding[1] is adopted by the other systems. In a near-instantaneous companding system, all samples in successive blocks of digital samples, typically 32 samples long, are coded to an accuracy determined by the size of the largest sample in each block. A separate scale-factor word is transmitted to inform the receiving terminal of the coding accuracy for each block of samples. Fig. 2 shows the compression characteristics of a 4-range near-instantaneous companding system. The samples in each block are coded for transmission according to the characteristic with the steepest slope which can accommodate the largest sample within the coding range.

The BBC NICAM-1 system uses four coding ranges with 13-bit initial coding accuracy and subsequent compression to 10 bits.[2] The TDF 9-bit system encodes each sample with linear 14-bit p.c.m., and the sample-words are then compressed to 9 bits by means of a 6-range law.

All three systems mentioned so far are intended for coding 6 high-quality sound programme channels into a

2,048 kbit/s digital bit-stream. The importance of 2,048 kbit/s is that it is the first-order multiplex bit-rate chosen by the UK Post Office, and data transmission capacity for this standard will be available in most parts of the UK when plans for the Post Office digital communications network have been implemented. The European postal and telecommunication authorities will also adopt the 2,048 kbit/s bit-rate as a standard. The normal function of a 2,048 kbit/s circuit will be to carry 30 telephone channels (64 kbit/s each), with two further 64 kbit/s channels used by the telecommunications authority for framing and signalling.

The A-law system as currently developed occupies the entire 2,048 kbit/s circuit capacity for its six programme channels, as does NICAM-1. Signals for frame synchronisation, error protection and supplementary low bit-rate communication are included in the 2,048 kbit/s 'package'. The 6-channel TDF 9-bit system occupies only thirty of the thirty-two 64 kbit/s channels, leaving 2 channels free for the postal and telecommunications authority to use for their own framing and signalling, as is normal when the circuit is in use for telephone traffic. An important feature of the TDF system is that each programme channel with its own frame code and error protection has a bit-rate which can be accommodated in five 64 kbit/s telephony channels so that partial use of the 2,048 kbit/s circuit is possible, if the full 6 programme channels are not required, and the digital sound-programme signals could therefore share the data circuit with other traffic.

With the possibility of partial access in mind, a new BBC NICAM system was considered, NICAM-2, which uses 14-bit initial coding and compression to 11 bits/sample. This system would offer the same flexibility as the TDF 9-bit system, with partial use of the 2,048 kbit/s circuit, and with two 64 kbit/s channels left clear for use by the Post Office. It would also have a lower level of idle-channel noise than NICAM-1 (owing to the use of 14-bit coding) and lower programme-modulated noise levels than any of the other companding systems, as it would use compression to 11 bits instead of 10 or less.

The total bit-rate requirements of each NICAM-2 programme-channel would be such that it could be accommodated in six telephony channels; the 2,048 kbit/s circuit would thus provide up to only 5 programme sound channels.

After the subjective tests had been conducted on the four systems described above, and the results studied, it appeared that the NICAM-2 system, though excellent in performance, would make rather extravagant use of bit-rate, reducing the programme-modulated noise to an unnecessarily low level. Another NICAM system, NICAM-3, was therefore proposed, using 14-bit initial coding with compression to 10 bits. It was felt that this system would be the optimum, having a similar programme-modulated noise to the already-accepted NICAM-1 system with a comparable bit-rate, but with a lower idle-channel noise (an aspect of NICAM-1 that has sometimes been criticised). Also its lower bit-rate in comparison with NICAM-2 would allow the use of 6 programme sound channels in a wholly dedicated 2,048 kbit/s circuit.

An extra subjective test was therefore run specifically to assess the performance of NICAM-3 in comparison with NICAM-1.

A summary of the main parameters of each of the five companding systems tested is given in the Appendix.

3. Experimental arrangements

3.1. The use of a micro-computer to represent a companding system

In order to avoid the need to construct p.c.m. equipment for each of the systems to be tested, a micro-computer* was programmed to perform the necessary digital compression and expansion functions in real time.[3] Thus it was possible to represent accurately the A-law, TDF 9-bit, NICAM-1, NICAM-2 and NICAM-3 systems by using a 14-bit a.d.c. at the input of the micro-computer and with its output connected to a 14-bit d.a.c.

When the micro-computer was required to reproduce the characteristics of the 13-bit NICAM-1 system, the a.d.c. and d.a.c. were switched to work as 13-bit devices, with dither signals added in the a.d.c.[4]

3.2. The preparation of a digital sound tape for subjective tests

It is likely that some of the longer programme circuits will use a combination of analogue and digital links, at least during the early development of digital networks. This means that the signals will encounter a number of coding and decoding processes (possibly up to four within the UK). Because of this, some subjective tests were carried out using four successive codecs (coders and decoders), so that the effect of cumulative impairments could be assessed.

All the material used in subjective tests was recorded,** as 14-bit p.c.m. sound signals, using a multi-channel digital tape recorder.[5] The micro-computer was programmed to perform a single companding action, and its 14-bit output was recorded on one tape channel, to provide the single-codec test material. The recorded digital signal was then replayed into the d.a.c., fed once more through the a.d.c. and micro-computer, and re-recorded on another channel. This re-recording operation was then repeated twice more so that a digital sound signal which had been processed by four coders and three decoders was recorded on the tape. For listening tests it was necessary only to replay the appropriate track on the recorder via the 14-bit d.a.c.

The experimental arrangement used for making the recordings and conducting the subjective tests is shown in Fig. 3. Nine audio channels were available on the digital recorder, and the following recordings were provided for

* The method of using the micro-computer, a Plessey Miproc, for audio processing was devised by G.W.W. McNally.

** Digital sound recordings for this work were prepared by F.A. Bellis and M.K.E. Smith.

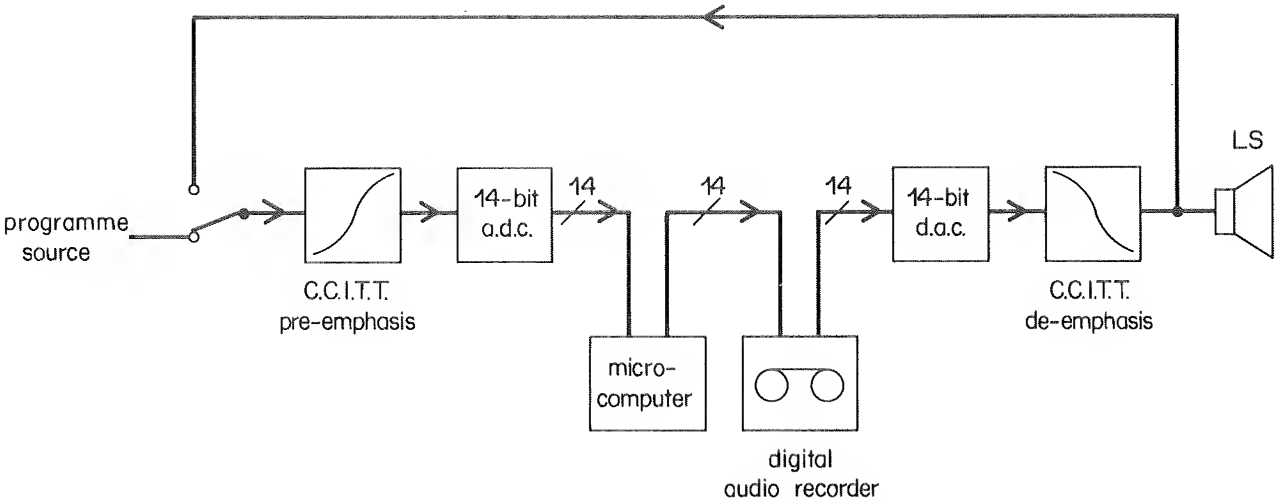

Fig. 3 - Arrangement for recording and subjective tests

the tests: A-law companding (single codec and four codecs in tandem), TDF 9-bit companding (single codec and four codecs), NICAM-1 (single codec and four codecs), NICAM-2 (single codec and four codecs) and 13-bit linear p.c.m. (single codec only). The 13-bit linear p.c.m. system, the coding system employed on the BBC's existing 13-channel p.c.m. distribution network, was included so that it could be used for comparison in some of the tests.

The analogue signal was subjected to CCITT pre-emphasis prior to each digital coding operation, and de-emphasis after each decoding operation. The attenuation of the pre-emphasis network was set to 6·5 dB at 800 Hz (at this setting the pre-emphasis gives a gain of 6 dB at 15 kHz).

In the preparation of the digital tape the signal levels were adjusted to peak at 6 on a peak-programme meter (corresponding to a signal level of +8 dBm) and no limiter was used. The test items were all EBU test material for which this adjustment was sufficient to ensure that no overloading occurred. Nevertheless, as a precautionary measure the test material was checked after CCITT pre-emphasis using a storage oscilloscope to verify that the peak signal level did not exceed the maximum for the a.d.c. (+9·5 dBm).

4. Description of tests

4.1. Measurement of signal-to-noise ratio

In order to check the operation of the micro-computer system, the ratio of peak signal amplitude to r.m.s. quantising noise at the output of the d.a.c. was measured for a single codec of each companding system without pre- and de-emphasis, over a wide dynamic range up to maximum signal level. The experimental arrangement used for making these measurements is shown in Fig. 4. Results of the measurements are shown in the graphs of Fig. 5.

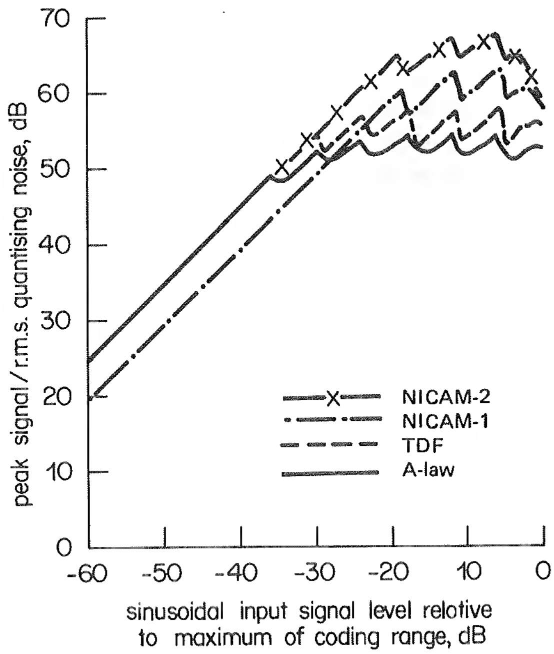

Fig. 5 - Measured ratio of peak signal to r.m.s. (unweighted) quantising noise for companding systems reproduced by the micro-computer

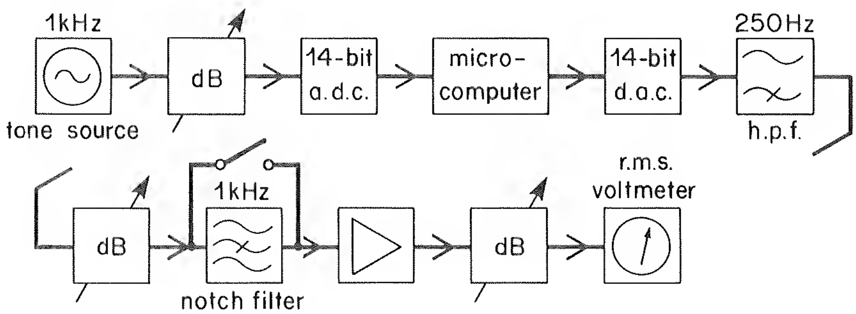

Fig. 4 - Arrangement for measurement of r.m.s. quantising noise

When measurements were made on the NICAM-2 system with a high level of input signal, the results obtained indicated that noise levels were somewhat greater than would have been expected from theoretical considerations. It is most likely that some slight intermodulation distortion in the analogue circuits of the d.a.c. and a.d.c. was responsible for this. The intermodulation products would have been at a sufficiently low level not to affect too seriously the measurements on the other companding systems, which were in reasonable agreement with theoretical predictions,[2] and it is unlikely that they would have influenced the results of subjective listening tests to any significant degree.

4.2. Subjective tests

Subjective tests were conducted in which listeners, comprising 17 technical staff who had some previous experience of listening tests, compared the performance of the companding systems. Some of the tests also included comparisons between companded p.c.m. and the 13-bit linear p.c.m., so that the results obtained could be related to the currently-accepted BBC standard for digital sound-programme links.

Because of the relatively large number of tests to be carried out it was considered advisable to conduct them in two groups. In this way, listeners underwent tests for periods of only about twenty minutes at a time. The first group of tests involved comparison of A-law with TDF and TDF with NICAM-1 for single codecs and also for four consecutive codecs. In the second group of tests NICAM-1 and NICAM-2 were compared using single codecs and four consecutive codecs, and both of these systems were also compared with 13-bit linear p.c.m. (single codec only). Previous tests which had been conducted using commercial A-law equipment and the prototype NICAM-1 equipment had yielded results* for the A-law system in comparison with NICAM-1 and so there was no need to repeat this test. As NICAM-2 is an improved version of NICAM-1 it was decided that the most meaningful test would be to compare it with NICAM-1 and 13-bit linear p.c.m.

* These results have not previously been published.

Three musical passages were used for the tests; the electronic 'Frere Jacques' test signal,[6] an excerpt of Schubert piano music and a glockenspiel *arpeggio* (broken chord). The piano and glockenspiel items were taken from an analogue master tape recorded using Dolby-A noise reduction.

The tests were conducted in a listening room with a low ambient noise level and representative of good domestic listening conditions; the mid-band reverberation time was about 0·3 s and the room volume 85 cubic metres. The test material was replayed via a high-quality monitoring loudspeaker type LS5/6.

Listeners were instructed to compare two systems, A and B, which were presented twice in each test and in the sequence: ABAB, and to award the test a grade according to the CCIR 7-point comparison scale.* The attention of the listeners was drawn to the noise which accompanied the programme and an example of severe programme-modulated noise was demonstrated prior to the tests. Duplicated tests were included, with the systems being compared in the reverse order on the second test. Listening level was adjusted before the start of tests so that the highest peaks of the music reached 92 dBA.

After the main subjective tests, using the digital sound recorder, a further brief subjective test was conducted to compare a single codec of NICAM-1 with a single codec of the system devised during the tests, NICAM-3. NICAM-3 is, effectively, a compromise between NICAM-1 and NICAM-2 in that it has 14-bit initial coding (to maintain a

* The CCIR 7-point scale is as follows:

3. A much better than B	−1. A slightly worse than B
2. A better than B	−2. A worse than B
1. A slightly better than B	−3. A much worse than B
0. A same as B	

TABLE 1

Mean grades awarded by listeners in subjective tests comparing single codecs

Test Item: Systems Compared A	B	Electronic 'Frere Jacques' Test Signal	Piano Music (Schubert)	Glockenspiel Arpeggio	Mean Result for the 3 Items
A-law	TDF	−1·24 (0·18)	−0·35 (0·26)	−0·88 (0·24)	−0·82
NICAM-1	TDF	+1·29 (0·32)	−0·15 (0·27)	+1·24 (0·25)	+0·72
NICAM-2	NICAM-1	+0·32 (0·16)	−0·09 (0·17)	+0·47 (0·23)	+0·23
13-bit linear p.c.m.	NICAM-1	+0·71 (0·21)	+0·12 (0·21)	— —	+0·42
13-bit linear p.c.m.	NICAM-2	−0·32 (0·11)	−0·03 (0·21)	— —	−0·18
NICAM-3	NICAM-1	0 (0·16)	−0·22 (0·26)	+0·03 (0·18)	−0·06

Note: The standard error of the mean grade is given in brackets.

TABLE 2

Mean grades awarded by listeners in subjective tests comparing 4 codecs in tandem

Test Item / Systems Compared A	B	Electronic 'Frere Jacques' Test Signal	Piano Music (Schubert)	Glockenspiel Arpeggio	Mean Result for the 3 Items
A-law	TDF	−1·76 (0·14)	−0·12 (0·26)	−0·26 (0·33)	−0·71
NICAM-1	TDF	+2·24 (0·26)	−0·41 (0·24)	+1·88 (0·32)	+1·24
NICAM-2	NICAM-1	+0·88 (0·36)	+1·24 (0·25)	+0·65 (0·26)	+0·92

Note: The standard error of the mean grade is given in brackets.

low level of idle-channel noise) but with compression to 10 bits/sample so that bit-rate requirements are lower than for NICAM-2. For this test, therefore, the micro-computer was used on its own (i.e. without the digital sound recorder) to reproduce the effect of processing by the two NICAM systems, and ten of the listening panel were used to judge the results.

5. Discussion of subjective test results

The results of the subjective tests on the first four companding systems are shown in Tables 1 and 2. Table 1 gives the mean subjective grades obtained when single codecs of A-law, TDF and the two NICAM systems were compared with each other and with 13-bit linear p.c.m. The results of the comparisons between 4 codecs of A-law, TDF and the two NICAM systems are given in Table 2.

It is evident, from these tests, that NICAM-2 emerges as the system giving the lowest impairment levels, followed by 13-bit linear p.c.m. coding, NICAM-1, TDF and then A-law. The differences found between NICAM-2, 13-bit p.c.m. and NICAM-1 were relatively small; on average, and with single codecs, the results for these systems differed by less than 0·5 grade. With four codecs in tandem, the mean difference between NICAM-2 and NICAM-1 was less than 1 grade.

The overall result of the additional test was that no significant difference could be discerned between NICAM-1 and NICAM-3 for single codecs. This result is not particularly surprising as, with compression to 10 bits, the programme-modulated noise would be very similar with both systems, and the reduction in idle-channel noise afforded by the use of 14-bit coding in NICAM-3 will have been concealed by noise from the analogue master tapes in most of the tests.

6. Conclusions

Tests conducted on four digital sound companding systems have shown that the BBC NICAM-1 system is preferred to the French TDF 9-bit system and the Italian A-law system. They have also confirmed the result from previous work that NICAM-1 is only slightly inferior to 13-bit linear coded p.c.m.

A derivative of the original NICAM-1 system, NICAM-2, gave somewhat better results (and was slightly preferred to 13-bit linear coding, but would make rather extravagant use of bit-rate in order to reduce programme-modulated noise to an unnecessarily low level. Largely as a result of the main programme of tests the NICAM-3 system was evolved, using 14-bit initial coding with compression to 10 bits so that the idle-channel noise level would be lower than that obtained with 13-bit coding systems.

The subjective tests did not reveal a preference for NICAM-3 over NICAM-1, because the presence of background noise on the tape recordings used for the test material masked the idle-channel noise of the coding systems. (Also, the listeners had been asked to concentrate their attention on programme-modulated noise.) Nevertheless the adoption of NICAM-3 with the lower idle-channel noise resulting from the 14-bit initial quantising is felt to be justified. This aspect of its performance is compatible with the rest of the transmission chain, with modern good quality domestic receivers and with the digital recording and associated digital techniques which will eventually be used in studio centres.

As a result of the work described in this Report, therefore, it was decided that BBC development should be based on NICAM-3.

7. References

1. OSBORNE, D.W. 1972. Digital sound signals: further investigation of instantaneous and other rapid companding systems. BBC Research Department Report No. 1972/31.

2. OSBORNE, D.W. *and* CROLL, M.G. 1973. Digital sound signals: bit-rate reduction using an experimental digital compandor. BBC Research Department Report No. 1973/41.

3. McNALLY, G.W.W. 1978. The use of a programmed computer to perform real-time companding of high-quality sound signals. BBC Research Department Report No. 1978/14.

4. CROLL, M.G. 1970. Pulse code modulation for high-quality sound distribution: quantising distortion at very low signal levels. BBC Research Department Report No. 1970/18.

5. BELLIS, F.A. 1976. A multichannel digital sound recorder. *IERE*, 1976, Video and Data Recording Conference, Proc. No. 35, pp. 123, 126.

6. MANSON, W.I. *and* REID, D.F. 1977. Two audio test-signal generators for assessing programme-modulated noise in digital compandors. BBC Research Department Report No. 1977/4.

8. Appendix

A summary of the main characteristics of the companding systems tested is given below:

Name of System	Type of Companding	Coding Accuracy		No. of bits/ Sample After Compression	Bit-Rate Requirements	Further Information
		For Low-Level Signals	For High-Level Signals			
A-law	Instantaneous	14 bits/sample	8 bits/sample	10	2048 kbit/s for 6 channels	A-law, with only 8 bits/ sample coding for high-level signals, has highest level of programme-modulated noise.
TDF 9-bit	Near-instantaneous	14 bits/sample	9 bits/sample	9	1920 kbit/s for 6 channels or 320 kbit/s per single channel.	Lower programme-modulated noise than A-law. Uses 5 x 64 kbit/s telephony time slots per sound channel in 2048 kbit/s multiplex circuit.
NICAM-1	Near-instantaneous	13 bits/sample	10 bits/sample	10	2048 kbit/s for 6 channels	Lower programme-modulated noise than A-law or TDF 9-bit system. Initial coding accuracy of 13 bits/ sample gives higher idle-channel noise level.
NICAM-2	Near-instantaneous	14 bits/sample	11 bits/sample	11	1920 kbit/s for 5 channels or 384 kbit/s per single channel	Lowest programme-modulated noise of any system tested. Uses 6 x 64 telephony time-slots per sound channel in 2048 kbit/s multiplex circuit.
NICAM-3	Near-instantaneous	14 bits/sample	10 bits/sample	10	2028 kbit/s for 6 channels or 338 kbit/s per single channel.	Same level of programme-modulated noise as NICAM-1, but lower idle-channel noise. Up to 6 channels can be used, with partial access on dedicated 2048 kbit/s circuits (i.e. circuits not shared with other users), multiplex framing and justification using the extra 20 kbit/s. On shared circuits, partial access can be used to provide up to 5 mono sound channels.

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